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A NOVEL HIGH QUALITY FM STEREO DEMODULATOR USING HYBRID DIGITAL/ANALOGUE TECHNIQUES

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INTRODUCTION

FM Stereo broadcasting offers the potential for high quality sound from either Live broadcasts or recorded material. In practice however, its performance can be limited by noise, interference and circuit imperfections at the receiver. The advent of digital recording technology has increased the quality of the source material and has thus put further demands on the quality of the receiver. Also the possibility of more independent FM radio transmitters requires the receiver to perform well with co-channel interference.

An ideal FM stereo demodulator would have the following characteristics:

1. Low Distortion.
2. Good resistance to interference.
3. Low spurious noises.
4. Low noise.
5. Capability for integrated circuit implementation.

The rest of this paper describes a recently developed technique of stereo FM demodulation which, by integrating the FM demodulator and stereo decoding functions, can better achieve the above aims compared with current techniques. The paper will first briefly review the current techniques of FM stereo demodulation and its weaknesses; it will then describe the the new technique. The paper will then conclude with suggestions for further improvements to the circuit.

THE CURRENT TECHNIQUES OF FM STEREO DEMODULATION

Fig. 1 shows a typical FM receiver architecture. The receiver is a single conversion superhetrodyne with an intermediate frequency of 10.7MHz. The FM signal is demodulated, after limiting, using a quadrature demodulator and the stereo multiplex signal is then passed to a stereo demodulation chip which provides the desired Left and Right channel outputs.

The structure outlined above is ubiquitous, doubtless due to the ready availability of inexpensive integrated circuits for each of the functions outlined. However, the structure is not without problems.

The first is the fact that the quadrature demodulation system (fig 2) using tuned circuits is inherently non-linear and so

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causes distortion. This distortion may be reduced by using double tuned circuits, or delay lines, however it is still inherent in the system. It also requires a complex setting up procedure in production if the highest quality is to be achieved, and one questions whether the adjustments made remain accurate over the lifetime of the equipment. Several alternative techniques of FM demodulation have been proposed [Refs 1, 2]; however they have failed to gain general acceptance. This is probably due to the fact that the quadrature system is easily integrated and the other systems when used in conjunction with current FM stereo decoding techniques offered little advantage.

There are two ways of decoding the stereo multiplex signal, these are shown in figs. 3 and 4. They both use a phase locked loop technique for recovering the suppressed 38KHz subcarrier from the 19KHz pilot tone. However they differ in the technique they use to recover the stereo information.

The first and most popular technique is the switching demodulator. This technique treats the input signal as a time division multiplex one and recovers the desired left and right channels by gating the input signal between two outputs, L and R, under the control of the recovered 38KHz subcarrier. This technique has the advantage that it is easy to realise. Also because low distortion switches are easy to implement the technique has low distortion. Because the technique uses a 1:1 mark space ratio waveform one can show [ref. 3] that the separation is poor; each channel containing approximately 1/5 of the other. However by subtracting the same proportion of the interfering channel from the output signal it is possible to compensate for this effect. The technique requires little filtering and is easily integrated.

The main disadvantage of this technique is the need for a square wave switching signal. On closer analysis the effect of this switching signal is to multiply the input signal by its harmonics. The fundamental of 38KHz yields the desired stereo demodulation. However, the 3rd, 5th and greater harmonics merely make the circuit sensitive to interference at those frequencies (114KHz, 190KHz etc.). This means that any interference present in the demodulator output within 15KHz of those frequencies will produce spurious tones and noises in the output. Signal may be present at these frequencies due to the presence of a co-channel signal or they may be present due to self noise in the IF and FM demodulator circuitry prior to the multiplex decoder. One can add additional filtering to reduce this problem but the filters amplitude and group delay must be flat up to 57KHz if the stereo separation is to be maintained.

Another solution is to use a multi-level switching waveform [ref. 4] which has the characteristic of having the first few harmonics after the fundamental set to zero, a 3 level waveform can have harmonics 2, 3 and 4 at zero. This eases the specification required of any decoder pre-filtering but does increase the complexity of the switching circuitry.

The second technique treats the input signal as a frequency division multiplex signal and explicitly separates and then demodulates the double sideband suppressed carrier difference signal. The desired left and right channels are then obtained via a sum and difference matrix circuit. This technique can avoid the problem of spurious responses if a linear multiplier, and a 38KHz sine-wave reconstituted sucarrier, are used in the demodulation process. Unfortunately while a sine-wave subcarrier is easily supplied the design of a linear multiplication cell which does not add appreciable distortion to the signal is considerably more difficult. Also, the technique can require input filtering and maintaining the necessary delay and amplitude tolerances between the sum and difference signals can be difficult.

Both the above techniques can have a problem with spurious responses in the PLL used to extract the 19KHz pilot tone. Again a square wave is often used as the waveform to one side of the phase detector and this results in the loop being sensitive to signals at 57KHz, 95KHz etc. If noise or interference is present in the input at those frequencies then it will enter the loop and cause jitter in the recovered subcarrier. This will degrade the separation of the stereo signal.

It would appear that the ideal stereo decoder would combine the simplicity and low distortion of a switching type decoder with the freedom from spurious response of the direct demodulation type of decoder.

A NOVEL STEREO DECODER

One of the most linear forms of FM demodulator is the pulse counting discriminator. Its ability to provide a high linearity repeatedly in production often makes it the first choice for providing FM detection in modulation or spectrum analysers [refs. 5, 6]. The technique is very simple, the zero crossings of the FM waveform are used to provide the trigger for a pulse of a fixed voltage and length. Thus the FM signal is converted to a Pulse Duration Modulated (PDM) signal. This signal contains a baseband component which is the original modulating signal. This may therefore be recovered by simply low-pass filtering the PDM signal. Providing the conversion circuit produces an output pulse of consistent amplitude and time the system is highly linear. When this technique has been used before the stereo demodulation function has been considered separately. Thus the pulses have been filtered to the normal stereo multiplex signal which is then decoded using one of the previously mentioned techniques.

However, consider the spectrum of the PDM signal. It can be shown [refs. 7, 8] that a PDM signal consists of three things; a

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DC component, the desired signal, and harmonics at multiples of the pulse repetition rate which have been phase modulated by the message signal. This is shown in fig. 5. Providing the IF is high enough the side-bands of the phase modulated components cause no problems as regards interference.

Fig. 5 shows that a two level pulse signal contains all the information necessary to decode the stereo multiplex signal. Therefore why bother converting it to a linear analogue signal prior to stereo decoding? Instead why not use this digital signal to control switches in such a way that when the output is filtered the desired left and right channels are obtained? This can be done by using the PDM signal to switch a sine-wave at the subcarrier frequency as shown in fig. 6. This has the effect of multiplying the input by the subcarrier and adding it to the baseband. The net result is that the output of the switch represents the left or right channels depending on the sign of the subcarrier input to the switch. One can then pass the output of the switch to a low-pass filter to reconstruct the desired analogue left and right channels.

The advantage of this technique is that the stereo demodulation does not introduce any sensitivity to spurious frequencies because the subcarrier is a sine-wave. However it still retains the advantages of the switching type decoder as regards ease of implementation and freedom from distortion. The system also reduces the filtering requirements of the conventional PDM system by using the necessary multiplex filters as the reconstruction filters as well.

The same technique can also be applied to the PLL used to extract the 19 KHz pilot tone with similar advantages. The complete system is shown in fig. 7 and is potentially easy to integrate.

FURTHER-WORK

The current implementation could be improved in order to make it easier to integrate or manufacture and also to improve its distortion performance.

Currently the necessary sine-waves are provided by dividing down a high frequency VCO and using a weighted resistor network to convert the counter outputs to a sine-wave. This technique requires odd resistor valves and is very sensitive to component tolerance. It would be much better to use a counter addressing a ROM which controlled a D/A convertor to provide the necessary sine-waves (see fig. 8). It can also provide the signals needed to implement a pilot cancellation scheme.

Just filtering a PDM signal does not give totally distortion free demodulation [ref. 7] and what one really needs to do is to convert the PDM to pulse amplitude modulation and then filter. This is easily done by using a switched current source to charge and discharge a capacitor. As current output digital to analogue converters are readily available, these can be combined with the first improvement.

The third and final improvement is to the FM-PDM conversion process. The whole system is critically dependent on the timing consistency of the output pulses. Currently we use RC timed monostables, however it would be better to use a ringing LC or crystal technique similar to that used by HP in their spectrum analysers [ref. 5]. This uses the ringing of a high Q resonator to control a counter in such a way as to provide very accurately timed pulses and would thus improve the distortion and noise performance of the system.

CONCLUSION

By considering the spectrum of the output of a pulse counting FM demodulator one has developed a novel integrated FM stereo demodulator/decoder which combines the advantages of the conventional approaches and offers improvements in the quality of the output signal. Furthermore the system could be easily fabricated on an integrated circuit. This offers the possibility of an I.C. which takes 10.7 MHz I/F in and produces high quality stereo as its output.

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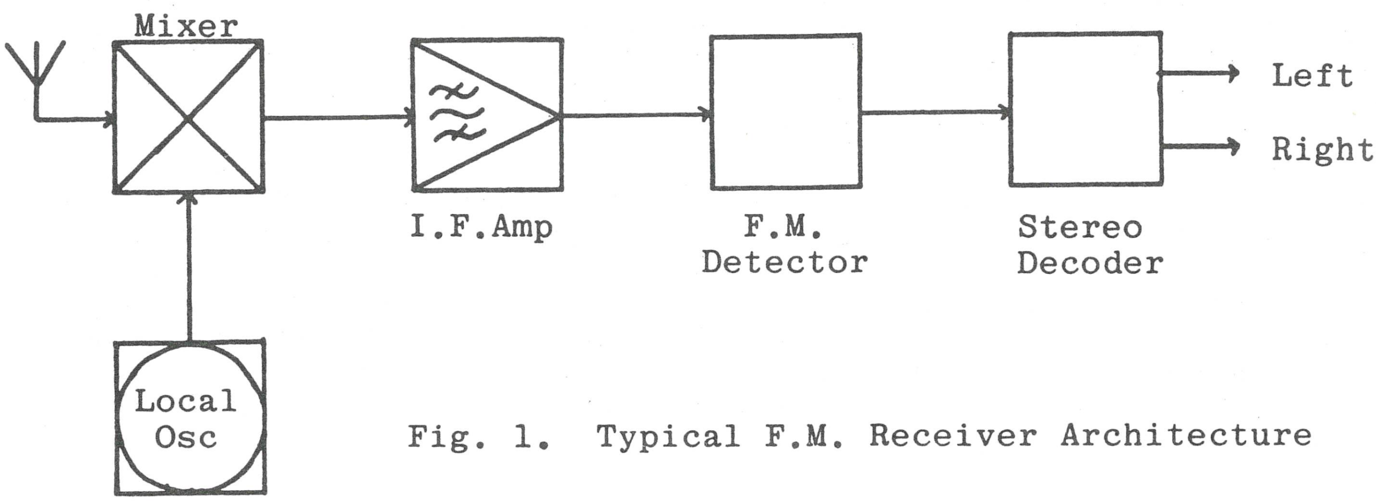

Fig. 1. Typical F.M. Receiver Architecture

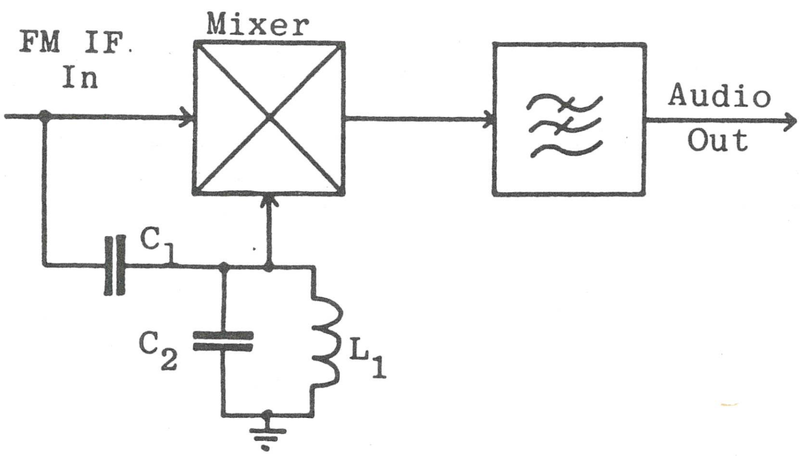

Fig. 2. Quadrature FM Demodulator

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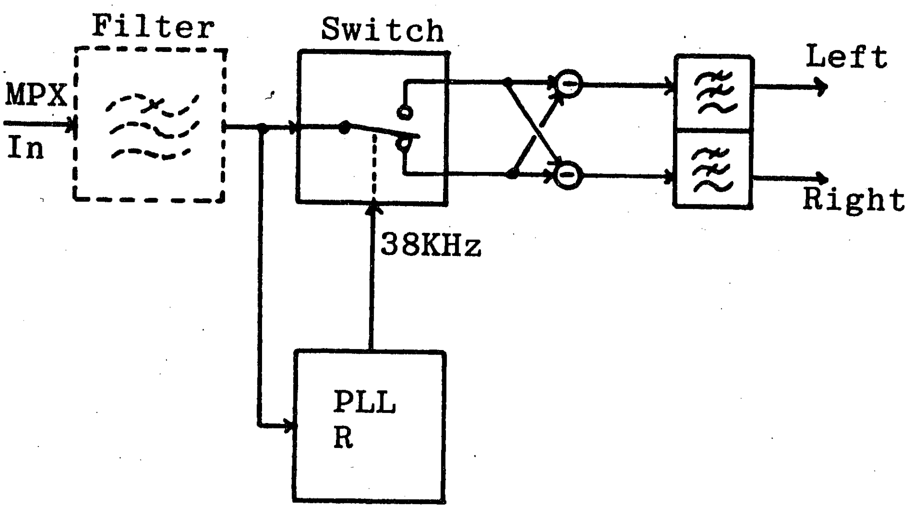

Fig. 3. Switching Stereo Decoder

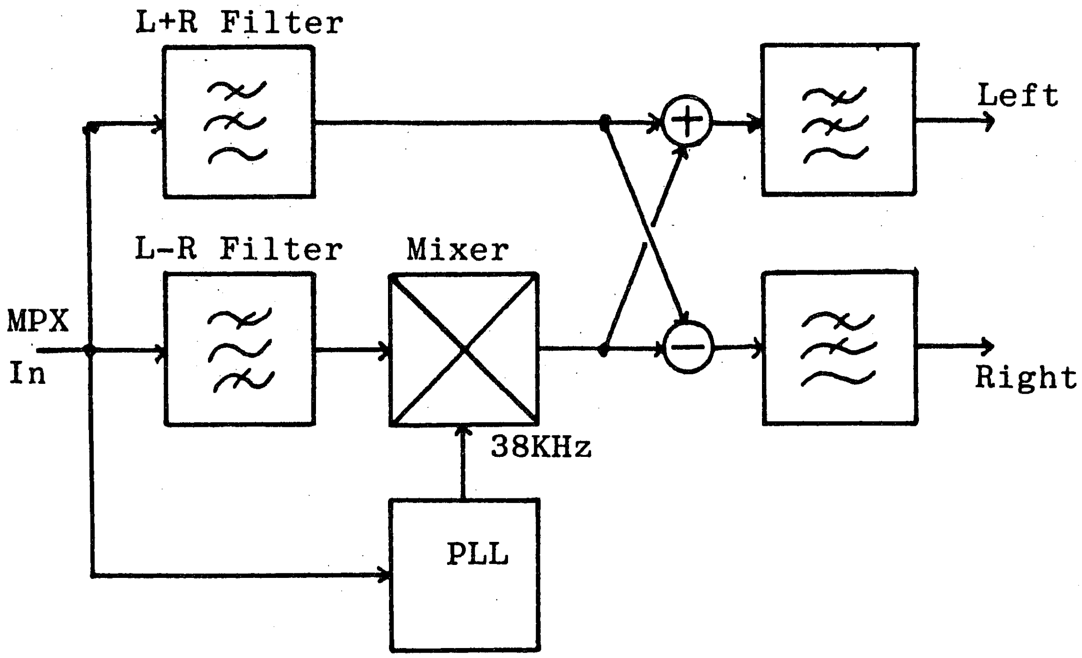

Fig. 4. Direct Demodulation Stereo Decoder

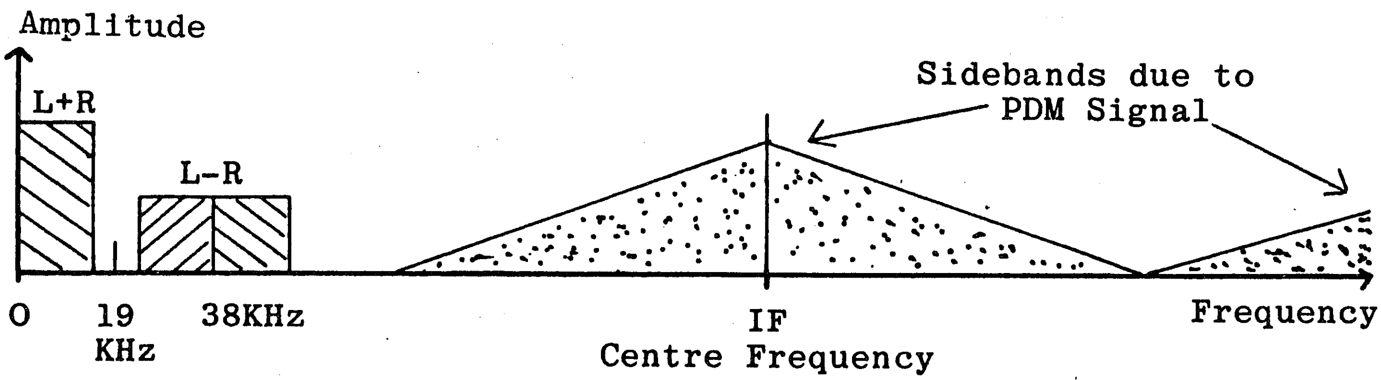

Fig. 5. Spectrum of a PDM FM Stereo Signal

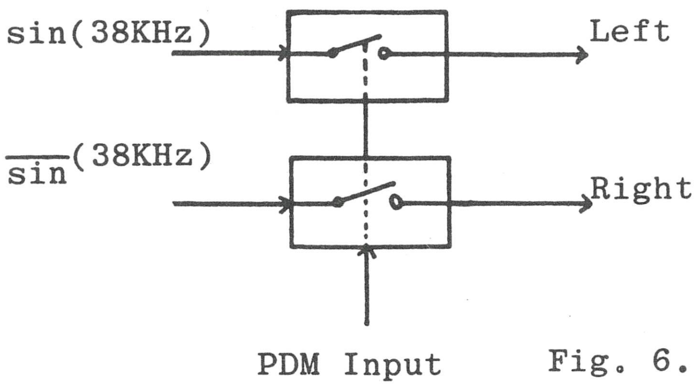

Fig. 6. Switching Demodulator using PDM

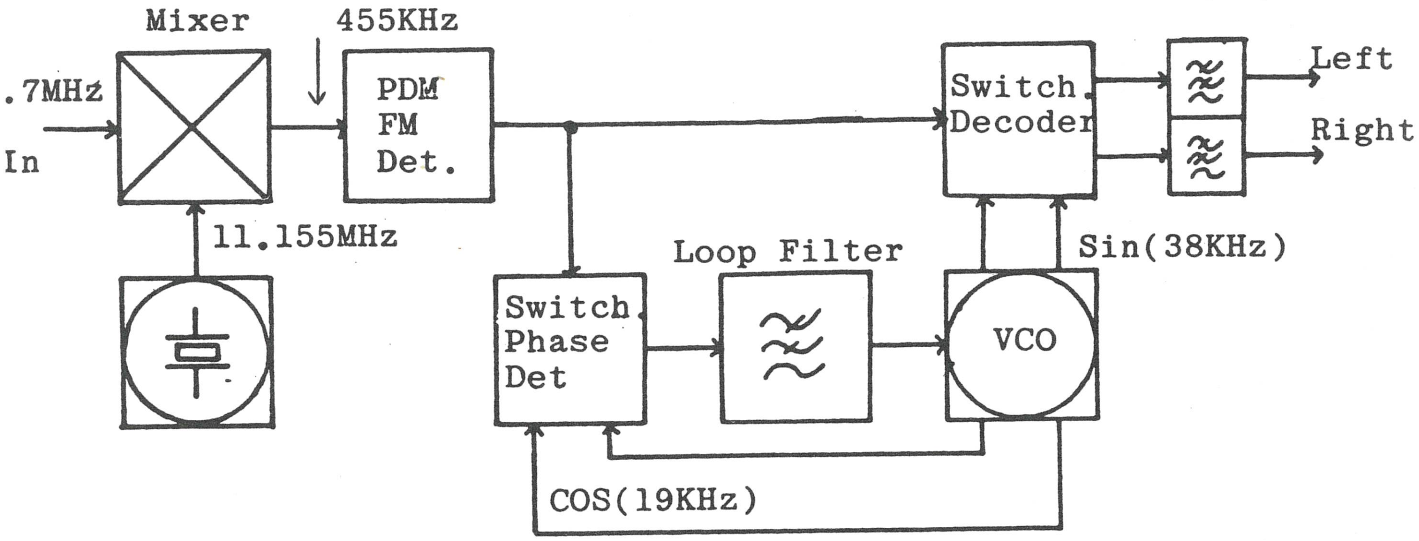

Fig. 7. Complete Hybrid Demodulation/Decoding System.

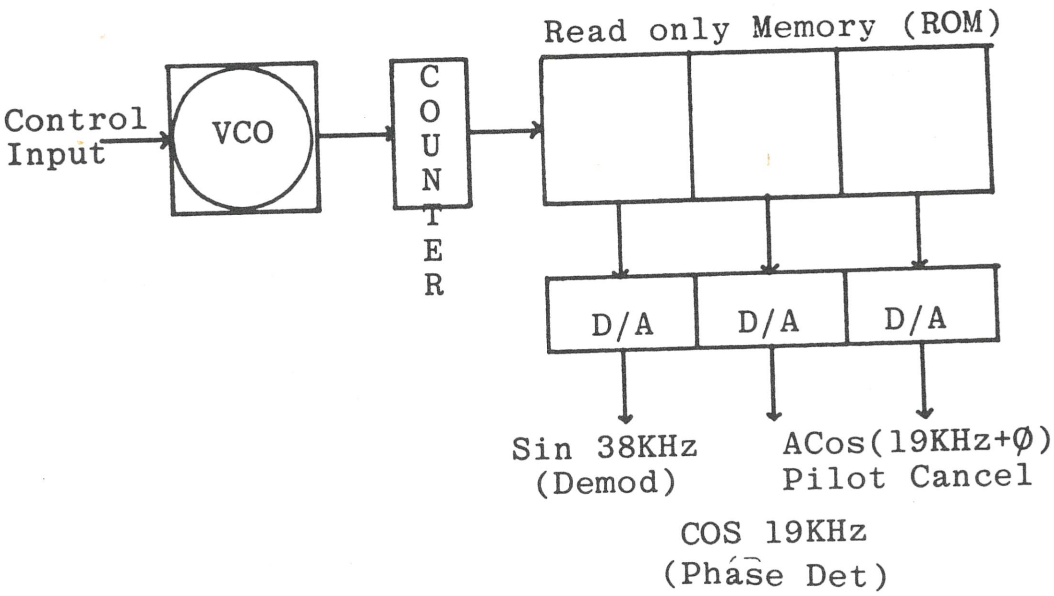

Fig. 8. Improved sine wave generation